Amendments to the Claims:

This listing of claims will replace all prior versions, and listings, of claims in the application:

Listing of Claims:

1. (Previously Presented) A method for performing call admission control, comprising:

(a) determining at least one of (i) a bandwidth utilization level for a first path including a first link, and (ii) an available bandwidth level for the first path and one or more Quality of Service or QoS metrics for the first path;

(b) comparing the determined at least one bandwidth level and the one or more Quality of Service or QoS metrics to one or more selected thresholds to determine whether a new live voice communication may be set up with a first selected codec;

(c) when a new live voice communication may not be set up with the first selected codec, performing at least one of the following steps:

(i) selecting a second different codec from among a plurality of possible codecs for the new live voice communication, wherein the second codec has a lower bit rate than the first codec;

(ii) changing an existing live voice communication from the first codec to the second codec; and

(iii) redirecting the new live voice communication from the first path to a second different path, wherein the second path does not include the first link.

2. (Original) The method of claim 1, wherein substep (c)(i) is performed and further comprising:

receiving a request to place the live voice communication; and

setting up the live voice communication with the second codec.

3. (Currently Amended) The method of claim 2, wherein each of a plurality of codecs has a corresponding bit rate and/or required ~~bandwidth~~utilization threshold~~level~~ and the selecting step comprises:

comparing at least one of the available bandwidth level and the bandwidth utilization level with the plurality of bit rates and/or ~~bandwidth~~utilization thresholds~~levels~~; and

selecting the highest quality codec having a corresponding bit rate and/or ~~bandwidth level~~utilization level permitted by the at least one of the available bandwidth level and the bandwidth utilization level.

4. (Original) The method of claim 2, wherein the comparing comprises:

comparing at least one of (i) a bandwidth utilization level and (ii) an available bandwidth level with one or more selected thresholds; and

comparing one or more Quality of Service or QoS metrics with one or more selected thresholds, wherein the second codec has a bandwidth usage characteristic sufficient to satisfy the comparing steps.

5. (Currently Amended) The method of claim 4, wherein the comparing step (b) comprises:

(B1) ~~estimating an impact on~~adjusting the one or more QoS metrics [[from]]to reflect placing the new live voice communication with the ~~second~~first selected codec;

(B2) determining whether the adjusted one or more QoS metrics are acceptable in view of selected thresholds; and

(B3) applying the following rules:

(B3i) when the adjusted QoS metrics are acceptable, setting up the new live voice communication with the first selected codec; and

(B3ii) when the adjusted QoS metrics are not acceptable, performing step (c).

6. (Currently Amended) The method of claim [[2]]5, wherein, when there is no codec from among the plurality of codecs that satisfies the one or more thresholds, performing one or more of blocking the new live voice communication and redirecting the new live voice communication from a packet-switched network to a circuit-switched network.

7. (Original) The method of claim 1, wherein substep (c)(ii) is performed.

8. (Original) The method of claim 7, wherein, when the existing live voice communication was set up, the first and second codecs were identified as being acceptable to both endpoints.

9. (Original) The method of claim 7, wherein substep (c)(ii) comprises:

renegotiating with the destination the codec to be used in the live voice communication.

10. (Original) The method of claim 1, wherein substep (c)(iii) is performed.

11. (Original) The method of claim 1, wherein the first link corresponds to a first set of port numbers and the second link to a second set of port numbers, wherein the first and second sets of port numbers are non-overlapping, wherein packets addressed to one of the first set of port numbers are directed along the first link and packets addressed to one of the second set of port numbers are directed along the second link and wherein the redirecting step comprises:

selecting for the packetized live voice communication a port address from the first set of port numbers when a new live voice communication can be set up with the first selected codec and

selecting for the packetized live voice communication a port address from the second set of port numbers when a new live voice communication cannot be set up with the first selected codec.

12. (Original) The method of claim 1, wherein in the determining step the bandwidth utilization level is determined.

13. (Original) The method of claim 12, wherein the bandwidth utilization level is determined by polling a local edge router.

14. (Currently Amended) The method of claim 12, wherein the bandwidth utilization level is the end-to-end bandwidth determined using at least one of Reservation Protocol messages, Path Differentiated Services, and Multi-Protocol Lavel Switching.

15. (Currently Amended) The method of claim 1, wherein in the determining step the available bandwidth level is determined and compared with a maximum threshold provided by the equation: Maximum Threshold = Allocated Link VoIP Bandwidth – Bandwidth required for one VoIP call, where the allocated VoIP bandwidth is the bandwidth dedicated to VoIP calls and wherein, when the available bandwidth is below the maximum threshold, one or more of (i), (ii), and (iii) is performed.

16. (Previously Presented) The method of claim 15, wherein the available bandwidth level is determined by polling a local edge router.

17. (Currently Amended) The method of claim 15, wherein the available bandwidth level is the end-to-end bandwidth determined using at least one of Reservation Protocol messages, Path Differentiated Services, and Multi-Protocol Lavel Switching.

18. (Previously Presented) The method of claim 1, wherein one or more QoS metrics is determined for the first path.

19. (Original) The method of claim 18, wherein the one or more QoS metrics is at least one of packet delay, jitter, packet loss, the availability of Differential Services Code Point, and RSVP status.

20. (Original) The method of claim 15, wherein the available bandwidth level is the bandwidth allocated to live voice communications less the bandwidth utilization level.

21. (Currently Amended) A computer readable medium having processor executable instructions stored thereon that, when executed, perform the steps of claim 1.

22. (Canceled)

23. (Previously Presented) A call admission controller, comprising:

a processor operable to:

(a) determine at least one of (i) a bandwidth utilization level for a first path including a first link, and (ii) an available bandwidth level for the first path, and one or more Quality of Service or QoS metrics for the first path;

(b) compare the determined at least one bandwidth level and the one or more Quality of Service or QoS metrics to one or more selected thresholds to determine whether a new live voice communication may be set up with a first selected codec; and

(c) when a new live voice communication may not be set up with the first selected codec, perform at least one of the following operations:

(i) select a second different codec from among a plurality of possible codecs for the new live voice communication, wherein the second codec has a lower bit rate than the first codec;

(ii) change an existing live voice communication from the first codec to the second codec; and

(iii) redirect the new live voice communication from the first path to a second different path, wherein the second path does not include the first link.

24. (Original) The controller of claim 23, wherein operation (c)(i) is performed and further comprising the operations of:

receiving a request to place the live voice communication; and

setting up the live voice communication with the second codec.

25. (Currently Amended) The controller of claim 24, wherein each of a plurality of codecs has a corresponding bit rate and/or required ~~bandwidth level~~utilization threshold and the selecting operation comprises:

comparing at least one of the available bandwidth level and the bandwidth utilization level with the plurality of bit rates and/or ~~bandwidth level~~utilization thresholds; and

selecting the highest quality codec having a corresponding bit rate and/or ~~bandwidth level~~utilization threshold permitted by the at least one of the available bandwidth level and the bandwidth utilization level.

26. (Original) The controller of claim 24, wherein the comparing operation comprises:

comparing at least one of (i) a bandwidth utilization level and (ii) an available bandwidth level with one or more selected thresholds; and

comparing one or more Quality of Service or QoS metrics with one or more selected thresholds, wherein the second codec has a bandwidth usage characteristic sufficient to satisfy the comparing steps.

27. (Currently Amended) The controller of claim 26, wherein the comparing operation comprises:

(B1) adjusting~~estimating an impact on~~ the one or more QoS metrics [[from]]to reflect placing the new live voice communication with the ~~second~~first selected codec;

(B2) determining whether the adjusted one or more QoS metrics are acceptable in view of selected thresholds; and

(B3) applying the following rules:

(B3i) when the adjusted QoS metrics are acceptable, setting up the new live voice communication with the first selected codec; and

(B3ii) when the adjusted QoS metrics are not acceptable, performing the comparing operation.

28. (Currently Amended) The controller of claim [[24]]27, wherein, when there is no codec from among the plurality of codecs that satisfies the one or more thresholds, the processor one or more of blocks the new live voice communication and redirects the new live voice communication from a packet-switched network to a circuit-switched network.

29. (Original) The controller of claim 23, wherein operation (c)(ii) is performed.

30. (Original) The controller of claim 29, wherein, when the existing live voice communication was set up, the first and second codecs were identified as being acceptable to both endpoints.

31. (Original) The controller of claim 29, wherein operation (c)(ii) comprises:

renegotiating with the destination the codec to be used in the live voice communication.

32. (Original) The controller of claim 23, wherein operation (c)(iii) is performed.

33. (Original) The controller of claim 23, wherein the first link corresponds to a first set of port numbers and the second link to a second set of port numbers, wherein the first and second sets of port numbers are non-overlapping, wherein packets addressed to one of the first set of port numbers are directed along the first link and packets addressed to one of the second set of port numbers are directed along the second link and wherein the redirecting operation comprises:

selecting for the packetized live voice communication a port address from the first set of port numbers when a new live voice communication can be set up with the first selected codec and

selecting for the packetized live voice communication a port address from the second set of port numbers when a new live voice communication cannot be set up with the first selected codec.

34. (Original) The controller of claim 23, wherein in the determining operation the bandwidth utilization level is determined.

35. (Original) The controller of claim 34, wherein the bandwidth utilization level is determined by polling a local edge router.

36. (Currently Amended) The controller of claim 34, wherein the bandwidth utilization level is the end-to-end bandwidth determined using at least one of Reservation Protocol messages, Path Differentiated Services, and Multi-Protocol Lavel Switching.

37. (Currently Amended) The controller of claim 23, wherein in the determining operation the available bandwidth level is determined and compared with a maximum threshold provided by the equation: Maximum Threshold = Allocated Link VoIP Bandwidth – Bandwidth required for one VoIP call, where the allocated VoIP bandwidth is the bandwidth dedicated to VoIP calls and wherein, when the available bandwidth is below the maximum threshold, one or more of (i), (ii), and (iii) is performed.

38. (Previously Presented) The controller of claim 37, wherein the available bandwidth level is determined by polling a local edge router.

39. (Currently Amended) The controller of claim 37, wherein the available bandwidth level is the end-to-end bandwidth determined using at least one of Reservation Protocol messages, Path Differentiated Services, and Multi-Protocol Lavel Switching.

40. (Original) The controller of claim 23, wherein one or more QoS metrics is determined for the first path.

41. (Original) The controller of claim 40, wherein the one or more QoS metrics is at least one of packet delay, jitter, packet loss, the availability of Differential Services Code Point, and RSVP status.

42. (Original) The controller of claim 37, wherein the available bandwidth level is the bandwidth allocated to live voice communications less the bandwidth utilization level.

43. (New) A method, comprising:

providing access to at least first and second wide area network links, the first network link being less expensive to use than the second network link;

monitoring the first network link for a bandwidth utilization level;

when the bandwidth utilization level on the first network link is more than a selected threshold, directing outgoing voice calls over the second network link; and

when the bandwidth utilization level on the first network link is less than the selected threshold, redirecting outgoing voice calls over the first network link.

44. (New) The method of claim 43, wherein a local edge router is configured so that packets with destination ports within a defined range are routed to the first network link and packets with ports in a second different range are routed to the second network link and wherein, when outgoing voice calls are to be directed to the first link, the first range is used for destination ports of outgoing packets, wherein, when outgoing voice calls are to be directed to the second link, the second range is used for destination ports of outgoing packets, and wherein the first and second ranges are not overlapping.